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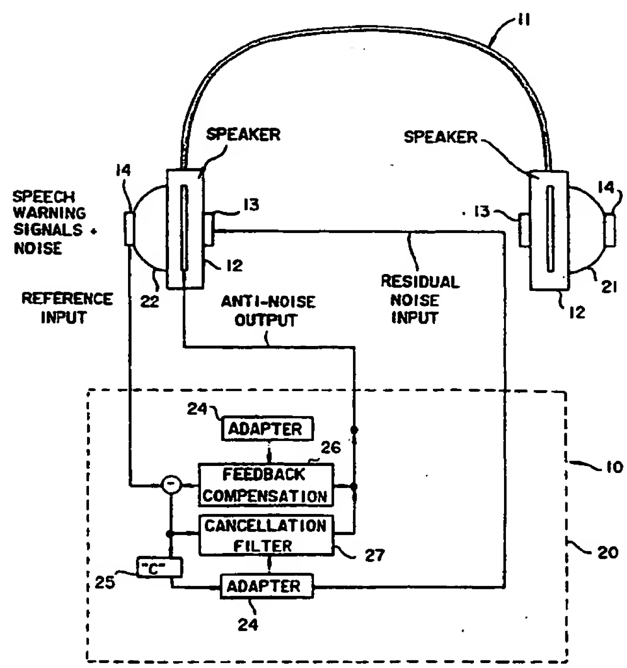
(54) Title: ACTIVE/PASSIVE HEADSET WITH SPEECH FILTER

(57) Abstract

An active/passive headset with a speech filter characterized by a controller (20) processing speech, noise and warning signals input from microphones (14) on closed back muffs (20) and providing an anti-noise output by a digital virtual earth or adaptive feedforward algorithm to speakers (12) adjacent a user's ear.



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ACTIVE/PASSIVE HEADSET WITH SPEECH FILTER

This invention relates to the area of headsets worn by a user in specific environments and, in particular, those areas where the background noise must be abated while simultaneously allowing speech through.

Generally, environmental noise can be broken into several areas such as noises of short duration with varying spectral characteristics such as conversations, other long duration noise with fixed or relatively fixed spectral characteristics such as operating pumps, fans, engines etc., and noises of very brief duration with "spike" characteristics such as something rattling, etc.

The instant invention is intended for use in environments found in industry, such as those requiring attenuation of low frequency noise as well as noise covering the speech band (300 - 3300 Hz).

Passive headsets have been employed and work extremely well at higher frequencies above 1000 Hz but don't perform at all in the lower frequency ranges. In these lower ranges active cancellation has been shown to work extremely well and the provision of high levels of protection are afforded users in the 50 to 1000 Hz range.

The instant headset provides protection at both the high frequency range (above 1000 Hz) as well as the low range (50 to 1000 Hz).

In the past, attempts to combine the two protections, i.e., high and low frequency attenuation, has resulted in not only the noise being attenuated but also the speech that the wearer needs to hear. Some systems met only limited success with fixed or "near-stationary" noise but not with the other noise of either (a) varying spectral characteristics or (b) brief duration noises with "spikes". Examples of such a system is found in U.S. Patent No. 4,025,721, to Graupe et al and U.S. Patent 4,185,168 to Graupe et al.

The instant invention solves the problem now existant, that of <u>total</u> attenuation of the noise and speech, by providing a solution of an active/passive headset that can employ any of several selective algorithms such as that disclosed in U.S. Patent No.

5,091,953 to Tretter, hereby incorporated by reference herein. Alternatively, it can employ the algorithm disclosed in U.S. Patent No. 5,105,377 to Ziegler which is also incorporated herein by reference. In addition, it can employ other algorithms such as that disclosed in the application of Ziegler in U.S. Patent Application No. 07/421 759 which is hereby incorporated by reference herein.

The technique of actively controlling noise is disclosed in various patents to Barry Chaplin especially U.S. Patent 4,654,871 which is hereby incorporated by reference herein.

Accordingly it is an object of this invention to provide an active/passive headset that will attenuate a wide range of environmental noise while allowing speech to be heard.

It is another object of this invention to provide a headset that blocks unwanted noise but allows speech to pass through a speech filter or "in-wire" canceller and be combined with the output of an active noise controller.

Still another object of this invention is to combine passive quieting with active noise cancellation in a headset and allow for passage of speech with simultaneous quieting of environmental noise.

These and other objects of the invention will become readily apparent when reference is had to the accompanying drawings in which

Fig. 1 shows a typical active/passive headset system incorporating the instant invention.

Fig. 2 shows a typical active/passive headset system using a digital virtual earth controller.

Fig. 3 is a more detailed block diagram of the digital virtual earth and adaptive feed forward systems shown in Figures 1 and 2.

Fig. 4 is a detailed description of an adaptive speech filter system.

Fig. 5 shows an active/passive closed back headset that combines the cancellation system with the speech filter system.

In Fig. 1 there is shown an active/passive closed back headset system 10. It consists of a typical passive headset 11, loudspeakers 12 that drive the anti-noise and residual microphones 13 to sense any remaining noise near the ear and reference microphones 14 to send advanced information for feed forward approaches and a system controller 20 which synthesizes the anti-noise signal.

The headset shown has closed backs 21, 22 for passive attenuation. Without the speakers, microphones and system controller, this headset would be a typical passive hearing protector.

The system is designed to use various algorithms such as that of Ziegler in U.S. Patent 5,105,377 or an adaptive feed forward approach by using the reference sensor as an input to the cancellation algorithm. Both these algorithms use a reference signal as inputs. The digital virtual earth (DVE) algorithm develops a reference signal by subtracting an equalized version of its own anti-noise signal from the residual signal. The adaptive feed forward uses the reference microphone as its input and is very effective on complicated noise environments that are broadband and random in character. The Least Means Square (LMS) adapter 24 shown in Fig 1 are Filtered - X versions which have inherent compensation for the effects of the feedback delays around the loop. Box "C" at 25 is the impulse response of the entire cancellation system.

Feedback compensator 26 and cancellation filter 27 complete the component portions of the controller.

DVE is highly effective to use in simple noise environments having only a few harmonics even where the noise varies tremendously.

Actuators 12 of the headset are large enough to be capable of producing anti-noise at the same level as the noise to be canceled. They have little or no distortion and have a minimum of input-to-output delay as any delay in the feedback loop slows down the system adaptation rate.

Residual sensors 13 are typically small electret microphones mounted on the speaker frame near the ear. It must faithfully reproduce the sound that remains at the

ear after cancellation so that the controller can make further adjustments to the anti-noise signal.

Reference microphones 14 are small electret microphones attached to the outside of the headset at a distance from the ear canal. This reference microphone is used to provide advanced information about the noise. The higher the frequency of the noise the more advanced information is needed to effectively cancel the noise, however, the passive muff limits the need for cancellation above 1000 Hz.

Figure 2 shows a basic block diagram of a digital virtual earth system, phenomena (such as noise) is detected by a residual sensor which sends out a sensor signal. The sensor signal is affected by filters and other factors shown in Figure 3 before reaching the processor. The DVE processor produces an output cancellation signal y. This output signal is affected by filters, and other factors. The output from the actuator combines with the original phenomena and the residual is detected at a location by the residual sensor of Figure 2.

The processor output for a specific sample k are given by

$$\underline{x}_k = \underline{r}_k - \underline{y}_{k-1} . \underline{C}_k$$

$$Y_k = -\underline{A}_k r_k$$

where

y_k is cancellation output value

$\underline{y}_{k-1}$ is a vector of previous cancellation output values

$\underline{C}_k$ is a vector of filter coefficients approximating S x E where S is the impulse response of the system after the processor and E is the impulse response of the system prior to the processor where a x b is the convolution of a and b.

$\underline{x}_k$ is a vector of estimates of the noise

$\underline{r}_k$ is a vector of most recent values of the residual signal

$\underline{A}_k$ is a vector of cancellation filter coefficients and a.b is the product of the two vectors.

When an LMS algorithm is used the cancellation filter coefficients are adapted as

$$\underline{g}_k = \underline{C}_k \underline{x}_k$$

$$A_{k+1} = A_k + \alpha_k g_k$$

where

$\underline{g}_k$ is the result of convolving $\underline{C}_k$ and $\underline{x}_k$

A_{k+1} is a vector of the most recent values of g_k and α is set for convergence.

Figure 3 shows in more detail the active cancellation system in either a DVE or AFF configuration. The active cancellation system has an input that receives residual noise signals (and reference signals in an AFF configuration) from a residual sensor (and reference sensor in the AFF configuration) and provides an output signal to an actuator. The actuator produces canceling phenomena from an electronic waveform (the output signal) into the area to be controlled (the ear canal). The canceling phenomena combines with the original phenomena so that the two sum algebraically.

Figure 4 shows in detail the adaptive speech filtering system. The system has as an input a reference signal that is composed of noise and speech from a reference sensor. From this input the adaptive speech filter separates the speech from noise and provides an output signal composed of speech only to an actuator.

Fig. 5 shows the instant invention with an adaptive speech filter 30 and the impulse "C" using a DVE cancellation approach. Reference microphones 14 not only pick up damaging noise but also pick up speech and warning signals. The embodiment of Fig. 2 is designed to provide protection against damaging noise which typically interferes with the speech band. It also attenuates noise in the audible range of the human ear effectively attenuating any warning siren. Using reference microphones 14 as inputs to a speech filter 30 that is then mixed with the anti-noise signal to be sent to the speaker is a means of allowing speech to pass through to the worker. If warning signals are outside the speech band and outside the frequency range of the active controller 20, these can be passed to the worker as well. Fig. 5 is a diagrammatic view

of the integration of active/passive hearing protection with speech (and warning signal) filtering so that local communication can be maintained for the safety of the worker.

The speech filter 30 separates speech from noise and output speech to the speaker with a minimal effect on intelligibility. The specific in-wire approach to use is dependent upon the noise but the DVE algorithm could be used. Various approaches have specific advantages in selectively filtering speech. Even the approach used in the Graupe patents can be used and since the output of filter 30 cannot interfere with the working of controller 20, it needs to be filtered from the control system as shown in Fig. 5 and, using the equations developed for the DVE algorithm, the output signal u_k is a specific sample k given by

$$w_k = \underline{T}_k \underline{v}_k$$

$$u_k = y_k + w_k$$

where

u_k is the cancellation and speech output value.

w_k is the filtered speech value.

$\underline{T}_k$ is a vector of filter coefficients determined by the adaptive speech filtering algorithm.

y_k is the cancellation output value as before.

$\underline{v}_k$ is the reference signal (speech and noise) input vector.

The residual sensor used to control the cancellation system also picks up the speech that is passed via the actuator. Therefore, the residual signal r_k must remove the speech signal w_k before input to the cancellation system so as not to interfere with the operation of the controller. It does so by modifying the residual signal r_k by

$$z_k = r_k - \underline{C}_k w_k$$

where

z_k is the filtered residual input to the controller.

$\underline{r}_k$ is the original residual input.

$\underline{C}_k$ is a vector of filter coefficients that is the overall impulse response of the cancellation system, as before.

w_k is the filtered output speech signal that is output to the actuator with the cancellation signal y_k.

The need to remove $\underline{C}_k w_k$ from the residual signal may not be required for all applications, for example, a siren canceling headset system will not try and cancel speech while the siren is in operation.

While the preferred embodiment of the invention is described it will be obvious to those of ordinary skill in the art that many changes and modifications can be made without departing from the scope of the appended claims.

CLAIMS

What is claimed is:

1. An active/passive noise canceling headset for attenuating high and low frequency noise, said headset comprising

a headset means having closed back earpiece means adapted to prevent external noise from entering the ear through passive attenuation,

speaker means associated with said earpiece means and adapted to produce anti-noise of the same level as the noise to be canceled,

reference sensing means also associated with said earpiece means and adapted to produce advanced information about the noise to be canceled, and

controller means adapted to secure information from said reference sensing means and cause emission of anti-noise by said speaker means.

2. A headset as in claim 1 and including residual sensing means also associated with said earpiece means and adapted to produce a signal characterizing the sound remaining at the ear.

3. A headset as in claim 2 and including a feedback loop means connecting said residual sensing means to said controller means and whereby said controller means makes adjustments to the anti-noise in response to the sound sensed by said residual sensing means.

4. A headset as in claim 3 wherein said reference sensing means are electret microphones or other type of microphones.

5. A headset as in claim 3 wherein said residual sensing means are electret microphones.

6. A headset as in claim 1 and wherein said controller means contains an adaptive speech filter means adapted to attenuate disturbing noise in the audible range of the human ear yet leave the speech to the headset wearer intelligible.

7. The method of controlling the amount of noise interfering with speech, said method comprising sensing signal containing speech and noise and converting it to electronic signals, using "in-wire" control approach producing "anti-noise" to cancel said noise component of the signal.

8. The method of claim 7 and including the step of passively quieting the higher frequency portions of the noise to be canceled simultaneous with the production of anti-noise.

9. The method of claim 8 and including the further step of sensing the noise remaining adjacent the ear after cancellation by said anti-noise and changing the anti-noise production in response to the amount of remaining noise..

10. The method of claim 9 and including the step of mixing the signals from sensing the noise containing noise and speech with the signal producing the anti-noise thus allowing speech to pass to the ear.

11. The method of filtering speech from noise said method comprising sensing signal containing speech and noise and converting it to electronic signals, adaptively filtering the signal to separate the noise component from the speech component and producing a signal that contains only the speech component.

12.. The method of manual selection of either the method of claim 7 or the method of claim 8 or both based upon characteristics of the noise environment wherein

said manual selection is a switch that allows the user to select a preferred approach for allowing speech to be most intelligible.

13. The method of automatic selection of either the method of claim 7 or the method of claim 8 or both based upon the spectral and temporal characteristics of the noise environment wherein said automatic selection is achieved using a detection algorithm that continuously monitors the spectral and temporal characteristics of the noisy signal.

14. The method as in claim 10 wherein said selection of adaptive speech filtering occurs when the noise is near-stationary and of limited bandwidth.

15. The method as in claim 10 wherein said selection of in-wire control occurs when the noise is varying and tonal or broadband.

16. The method of manual selection of claim 12 wherein a setting to automatic selection can be made.

17. The method of manual selection of claim 12 wherein a setting to an off position can be made.

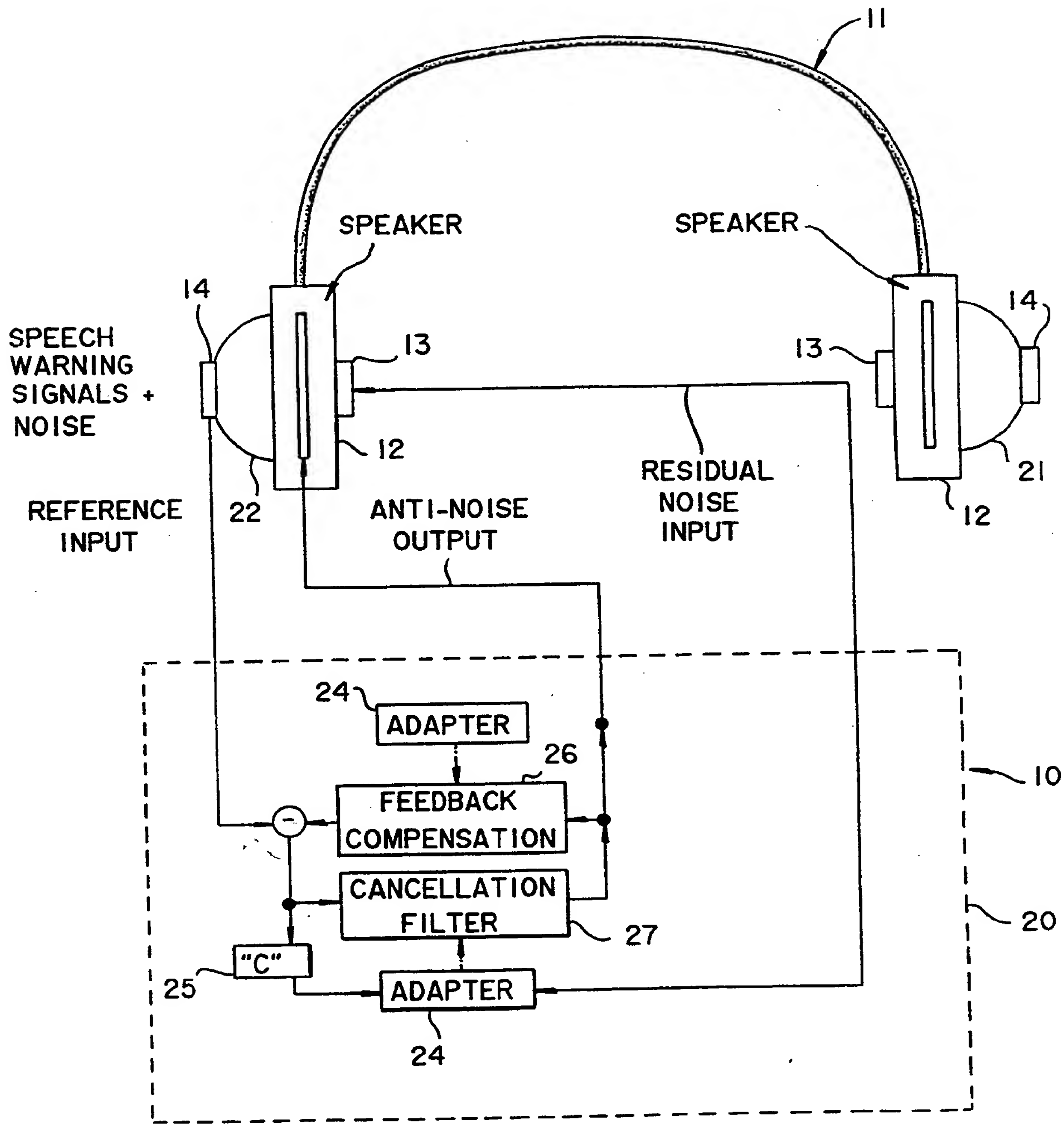
11
SPEAKER
SPEAKER
14
14
SPEECH
WARNING
SIGNALS +
NOISE
13
13
12
21
12
22
REFERENCE
INPUT
ANTI-NOISE
OUTPUT
RESIDUAL
NOISE
INPUT
24
ADAPTER
26
FEEDBACK
COMPENSATION
CANCELLATION
FILTER
27
"C"
25
ADAPTER
24
10
20

FIG.1

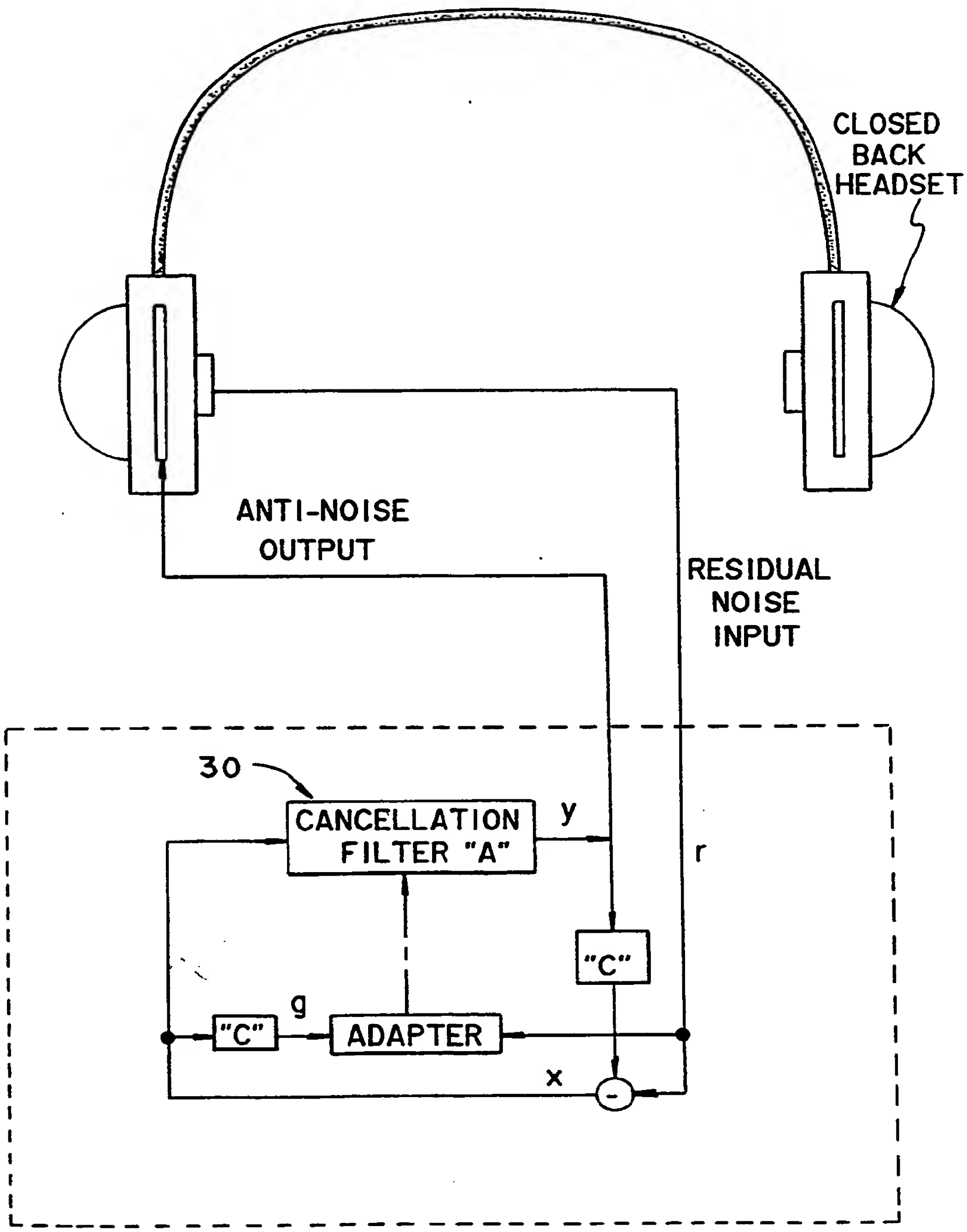
CLOSED BACK HEADSET
ANTI-NOISE OUTPUT
RESIDUAL NOISE INPUT
30
CANCELLATION FILTER "A"
y
r
"C"
g
"C"
ADAPTER
x
-

FIG.2

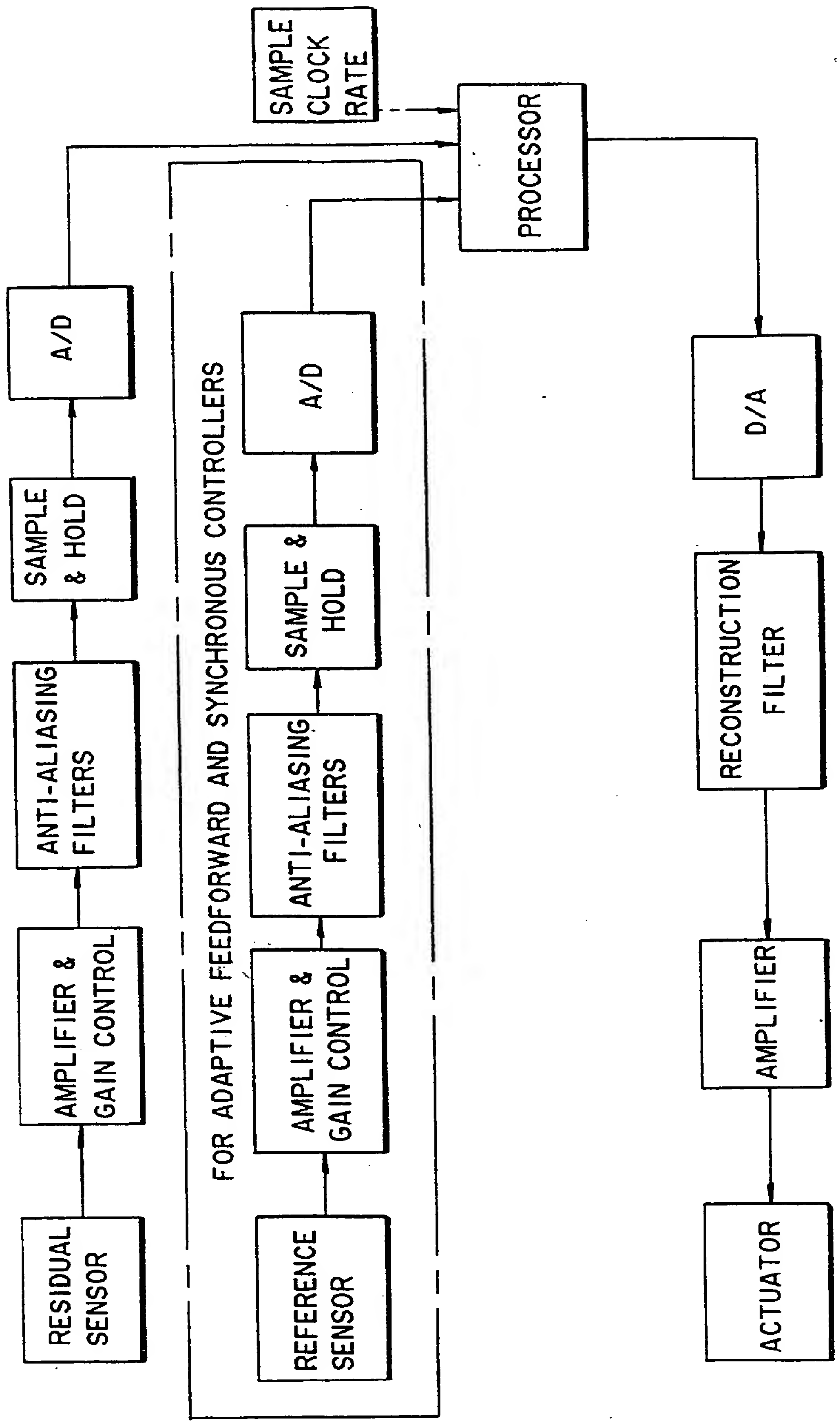
SAMPLE CLOCK RATE
PROCESSOR
FOR ADAPTIVE FEEDFORWARD AND SYNCHRONOUS CONTROLLERS
RESIDUAL SENSOR
AMPLIFIER & GAIN CONTROL
ANTI-ALIASING FILTERS
SAMPLE & HOLD
A/D
REFERENCE SENSOR
AMPLIFIER & GAIN CONTROL
ANTI-ALIASING FILTERS
SAMPLE & HOLD
A/D
D/A
RECONSTRUCTION FILTER
AMPLIFIER
ACTUATOR

FIG.3

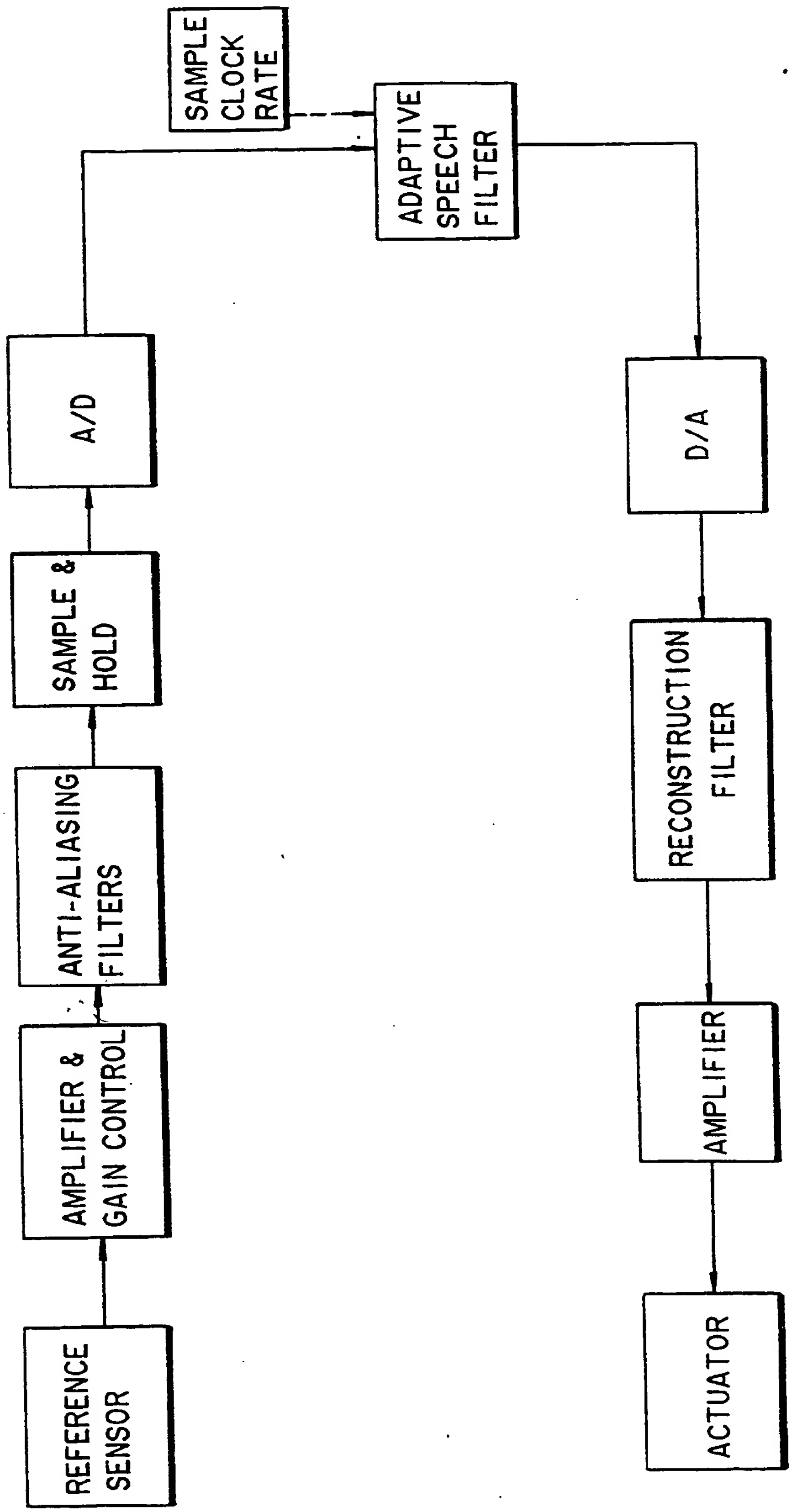
SAMPLE CLOCK RATE
ADAPTIVE SPEECH FILTER
A/D
SAMPLE & HOLD
ANTI-ALIASING FILTERS
AMPLIFIER & GAIN CONTROL
REFERENCE SENSOR
D/A
RECONSTRUCTION FILTER
AMPLIFIER
ACTUATOR

FIG.4

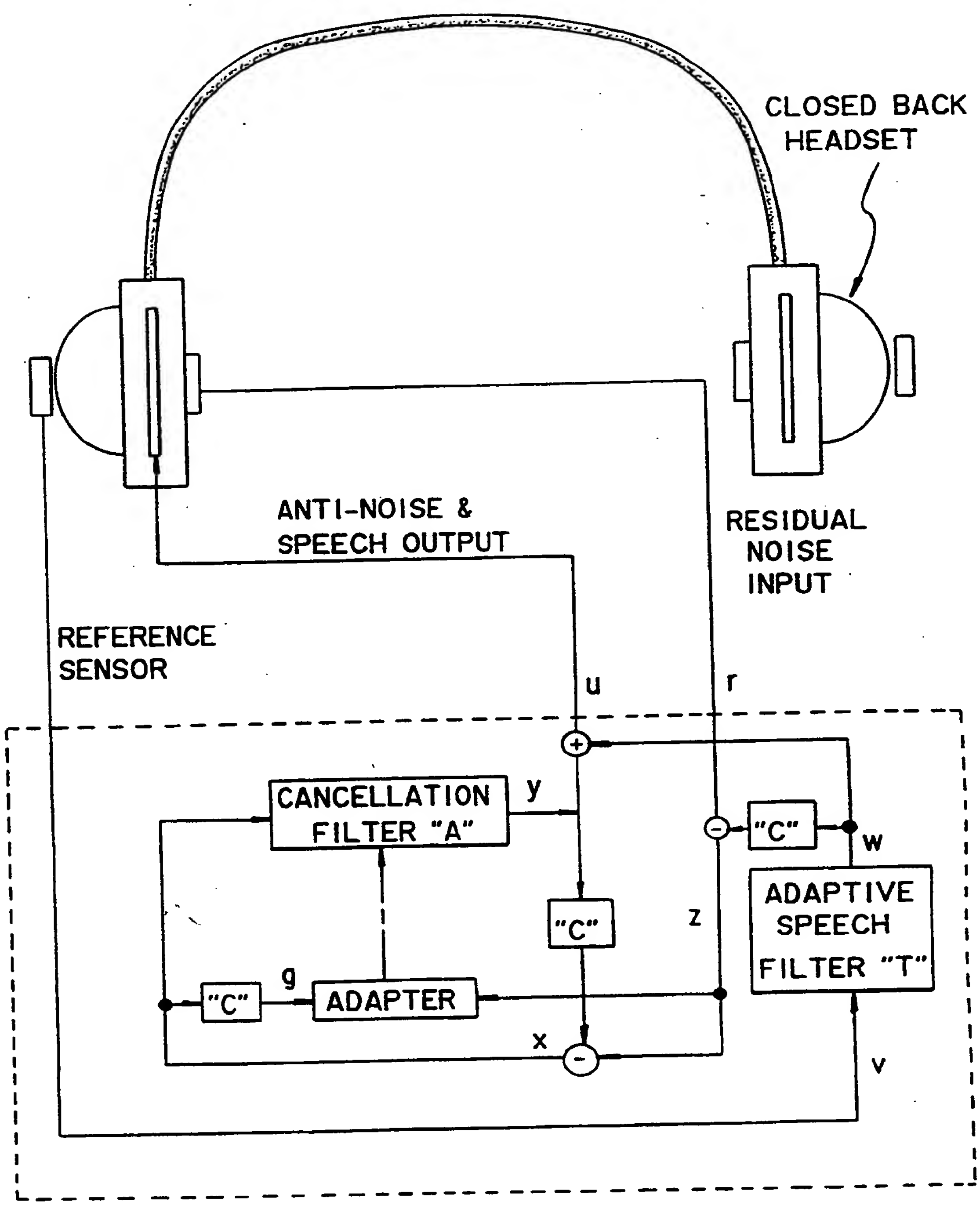
CLOSED BACK HEADSET
ANTI-NOISE & SPEECH OUTPUT
RESIDUAL NOISE INPUT
REFERENCE SENSOR
u
r
CANCELLATION FILTER "A"
y
"C"
w
ADAPTIVE SPEECH FILTER "T"
"C"
z
g
"C"
ADAPTER
x
v

FIG.5

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US92/04568

A. CLASSIFICATION OF SUBJECT MATTER

IPC(5) :H03B 29/00
US CL :381/71
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 381/72, 74

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X Y	US, A, 3,952,158 (Kyle et al.) 20 April 1976 See Figs. 1-5	1 1-6
X Y	US, A, 4,061,079 (Freifeld) 06 December 1977 See Figs. 2-5	1 2-6
Y,E	US, A, 5,126,681 (Ziegler, Jr. et al.) 30 June 1992 See Abstract and Figs. 1-3	7-17
Y	US, A, 5,105,377 (Ziegler, Jr.) 14 April 1992 See Abstract and Figs. 1-5	1-17
Y	US, A, 4,953,217 (Twiney et al.) 28 August 1990 See Abstract Fig. 1	2-6
Y	US, A, 5,091,953 (Tretter) 25 February 1992 See Abstract and Figs. 1 and 5	7-17
Y	US, A, 4,064,362 (Williams) 20 December 1977 See Abstract and Fig. 1	1
A	US, A, 4,677,678 (McCutchen) 30 June 1987 See Abstract and Figs. 1-7	1-6
A	US, A, 4,455,675 (Bose et al.) 19 June 1984 See Abstract and Figs. 1 and 5-6	1-6

[X] Further documents are listed in the continuation of Box C. [ ] See patent family annex.

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"&" document member of the same patent family

Date of the actual completion of the international search 06 AUGUST 1992	Date of mailing of the international search report 09 NOV 1992
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INTERNATIONAL SEARCH REPORT

International application No.
PCT/US92/04568

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US, A, 5,046,103 (Warnaka et al.) 03 September 1991 See Abstract and Figs. 1-2	1-6